Portable telephone set

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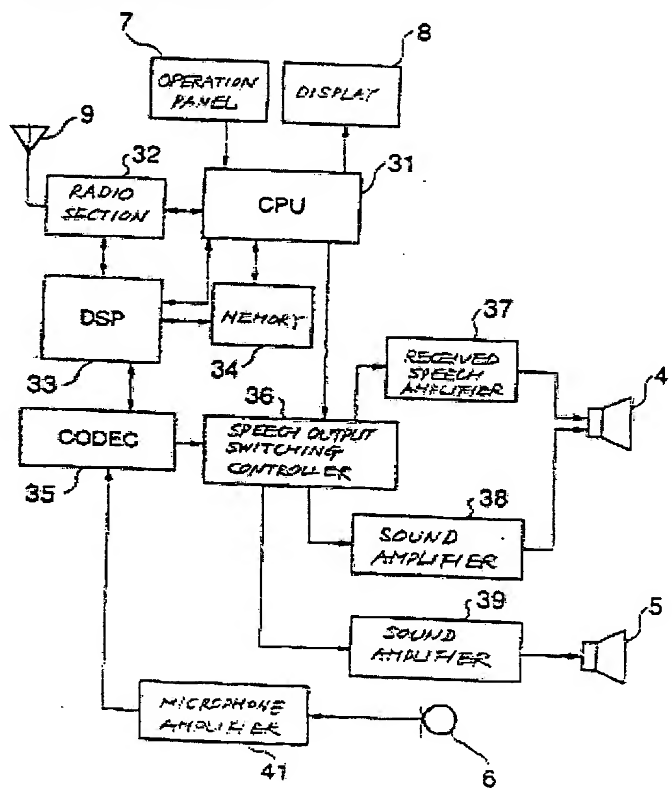
A cellular phone for interchanging information with a base station by radio of the present invention includes a first speaker for selectively outputting a received speech or sound and a second speaker for outputting sound. A controller controls the output of a received speech or sound from the first and second speakers in accordance with sound setting selected beforehand by the user of the phone.



Portable telephone set

Description of corresponding document: **US2002042287**

BACKGROUND OF THE INVENTION
[0001] 1. Field of the Invention
[0002] The present invention relates to a cellular phone for interchanging information with a base station included in a mobile communication system.
[0003] 2. Description of the Background Art
[0004] A modern cellular phone usually includes speakers for outputting a call incoming tone and a received speech and a memory for storing, e.g., call incoming tone data. More specifically, one of the speakers is assigned to a call incoming tone while the other speaker is assigned to a received speech and, in this sense, plays the role of a receiver. For the speaker assigned to sound, a speaker capable of outputting sound over a board frequency band is often used to meet the demand for a variety of call incoming tones.
[0005] In parallel with the increase in the functions available with a cellular phone, the capacity of the above-mentioned memory is increasing to such a degree that even music or speeches can be recorded in the phone. This allows the user of the phone to reproduce music distributed via a music distributing machine or Internet on the phone.
[0006] To allow the user of the cellular phone to reproduce music on the phone, the phone may be configured such that music based on music data is output from the speaker assigned to sound and higher in performance than the speaker assigned to a received speech. Such a configuration, however, does not provide the music with a stereophonic effect. While two speakers may be installed in the phone to output stereophonic sound, they increase the size and weight of the phone, which should be small size and light weight.
[0007] Technologies relating to the present invention are disclosed in, e.g., Japanese Patent Laid-Open Publication Nos. 1-120159, 4-243358, 6-37920, 10-23115, and 10-233826.

SUMMARY OF THE INVENTION
[0008] It is an object of the present invention to provide a cellular phone capable of outputting stereophonic sound while maintaining a small size, lightweight configuration.
[0009] A cellular phone for interchanging information with a base station by radio of the present invention includes a first speaker for selectively outputting a received speech or sound and a second speaker for outputting sound. A controller controls the output of a received speech or sound from the first and second speakers in accordance with sound setting selected beforehand by the user of the phone.

BRIEF DESCRIPTION OF THE DRAWINGS

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[0010] The above and other objects, features and advantages of the present invention will become more apparent from the following detailed description taken with the accompanying drawings in which:
[0011] FIG. 1 is an exploded isometric view showing a cellular phone embodying the present invention;
[0012] FIG. 2 is a schematic block diagram showing electric circuitry included in the illustrative embodiment;
[0013] FIG. 3 is a schematic block diagram showing a specific configuration of a speech output switching controller included in the circuitry of FIG. 2; and
[0014] FIGS. 3 and 4 are flowcharts demonstrating a specific operation of the illustrative embodiment.

DESCRIPTION OF THE PREFERRED EMBODIMENT
[0015] Referring to FIG. 1 of the drawings, a cellular phone embodying the present invention is shown and includes a circuit board 3. A first and a second speaker 4 and 5, respectively, a microphone 6, an operation panel 7 and a display 8 are mounted on the circuit board 3. The first speaker 4 selectively outputs a received speech or sound including a call incoming tone and music. The second speaker 5 outputs only sound. The first and second speakers 4 and 5 are spaced from each other on one major surface of the circuit board 3 so as to implement a stereophonic effect. The speaker 4 is positioned such that when the user of the phone converses on the phone, the speaker 4 faces the user's ear. During conversation, the user's voice is input to the microphone 6. The display 8 is implemented as, e.g., an LCD (Liquid Crystal Display) and displays various text information including phone numbers as well as graphic information. The user may operate the operation panel 7 in order to input various kinds of information including a phone number. Electric circuits for executing various operations required of the cellular phone are also arranged on the circuit board 3. The speakers 4 and 5 are equivalent in performance to each other.
[0016] A front case 1 is formed with holes 11, 12 and 13 corresponding in position to the speakers 4 and 5 and microphone 6, respectively. The front case 1 includes a transparent portion corresponding to the display 8. Further, a group of holes are formed in the front case 1 and correspond to a group of keys arranged on the operation panel 7. An antenna 9 is mounted on a rear case 2 and connected to the circuit board 3. The front and rear cases 1 and 2 are put together with the circuit board 3 intervening therebetween. At this instant, the surface of the circuit board 3 loaded with the various electric parts and circuits faces the front case 1.
[0017] FIG. 2 shows electric circuitry included in the illustrative embodiment. As shown, the circuitry includes a CPU (Central Processing Unit) 31 for receiving various signals including key input signals from the operation panel 7. The CPU 31 delivers a display command and a switch command to the display 8 and a speech output switching controller 36, respectively. Further, the CPU 31 interchanges information with a DSP (Digital Signal Processor) or signal processor 33 and a memory 34. The memory 34, which plays the role of a speech memory, may be implemented by an

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EEPROM (Electrically Erasable Programmable Read Only Memory) by way of example. In addition, the CPU 31 executes various kinds of control required of the phone. In the illustrative embodiment, the CPU 31 and speech output switching controller 36 constitute a controller in combination.
[0018] The DSP 33 interchanges information with the CPU 31 and memory 34 as well as with a radio section 32 and a CODEC (Coder/Decoder) 35. The CODEC 35 includes an AD (Analog-to-Digital) converter and a DA (Digital-to-Analog) converter. The memory 34 stores various data including call incoming tone data and is capable of storing other data including music data received via, e.g., Internet. The radio section 32 amplifies power and converts frequency in order to communicate with a base station, which is included in a mobile communication system, via the antenna 9. The DSP 33 corrects the characteristic of a received signal, sets up a radio channel between the cellular phone and a base station, switches a conversation channel, and processes a speech signal.
[0019] The speech output switching controller 36 is connected to a received speech amplifier 37 and sound amplifiers 38 and 39 as well as to the CPU 31 and CODEC 35. The switching controller 36 executes volume control and amplifier switching control in accordance with a command output from the CPU 31. The received speech amplifier 27 and sound amplifier 38 are connected to the first speaker 4 while the sound amplifier 39 is connected to the second speaker 5. The sound amplifiers 38 and 39 each have a greater amplification ratio than the received speech amplifier 37. More specifically, sound output from the speaker 4 via the sound amplifier 38 and sound output from the speaker 5 via the sound amplifier 39 each have a greater maximum volume that a speech output from the speaker 4 via the received speech amplifier 37. It is to be noted that a received speech refers to the voice of the user of another cellular phone communicating with the cellular phone of the illustrative embodiment.
[0020] The microphone 6 transforms a speech input thereto to a speech signal. A microphone amplifier 41 amplifies the speech signal output from the microphone 6 and feeds the amplified speech signal to the CODEC 35.
[0021] FIG. 3 shows a specific configuration of the speech output switching controller 36. As shown, the switching controller 36 includes a group of switches 311 through 313 and a group of volumes 321 through 323. The switches 311 through 313 and volumes 321 and 323 are generally designated by the reference numerals 310 and 320, respectively. The volumes 321 through 323 are electronic volumes whose resistance is variable under electric control.
[0022] The switches 311 and 312 are connected to one output of the CODEC 35 at one end thereof. The volume 321 has an input connected to the other end of the switch 311 and has an output connected to the input of the received speech amplifier 37. The volume 322 has an input connected to the other end of the switch 312 and has an output connected to the input of the sound amplifier 38. The switch 313 is connected to the other output of the CODEC 35 at one end thereof. The volume 323 has an input connected to the other end of the switch 313 and has an output connected to the input of the sound amplifier 39.
[0023] The switching controller 36 selectively turns on or turns off each of

the switches 311 through 313 and varies the resistance of each of the volumes 321 through 323 in accordance with a command output from the CPU 31. The CODEC 35 produces a particular signal from each of its two outputs when a stereophonic effect is desired or produces the same signal from the two outputs when such an effect is not desired.

[0024] In the illustrative embodiment, the speaker 4 is operable in two different modes, i.e., a received speech mode and a sound mode. Selection of one of these two modes will be referred to as sound setting. When the received speech mode is selected, the speaker 4 is caused to output a received speech with a speech volume set beforehand. When the sound mode is selected, the speaker 4 is caused to output sound with a preselected call incoming tone volume. As for sound setting, a particular mode, e.g., the received speech mode is set at the time of, e.g., shipment. The user of the phone can select a desired speech volume and a desired call incoming tone volume on the phone. Further, the user can select a desired pattern of call incoming tones. The speech volume refers to the volume of a received speech output via the speaker 4 while the call incoming tone volume refers to the volume of sound output via the speaker 5. When the user selects the sound mode available with the speaker 4, the speaker 4 is controlled to the same volume as the speaker 5. In the illustrative embodiment, the volume of the speaker 4 is sometimes raised to the volume of the speaker 5 stepwise, as will be described specifically later. The maximum volume of a call incoming tone, or sound, is greater than the maximum volume of a received speech.

[0025] The user can perform sound setting and select a desired speech volume and a desired call incoming tone volume when the phone is in a standby state or when conversation or music reproduction is under way. A specific operation of the illustrative embodiment will be described with reference to FIG. 4 as well as to FIG. 3. When the user presses a particular key, e.g., a menu key positioned on the operation panel 7, the CPU 31 displays a menu picture on the display 8. As shown in FIG. 4, the user, watching the menu picture, selects "sound setting" by moving a cursor by way of example (step S401). The user may additionally select a desired speech volume and a desired call incoming tone volume. The operation panel 7 delivers the results of user's selection to the CPU 31. In response, the CPU 31 determines whether or not the user has selected the sound mode available with the speaker 4 (step S402).

[0026] Assume that the user has selected the sound mode (YES, step S402). Then, the CPU 31 causes the speech output switching controller 36 to select a call incoming tone by feeding a command to the switching controller 36 (step S403). In response, the switching controller 36 turns off the switch 311 and turns on the switch 312 to thereby deliver the output of the CODEC 35 to the speaker 4 via the sound amplifier 38. In addition, the switching controller 36 controls the resistance of the volume 322 in order to implement the call incoming tone volume selected beforehand. The CPU 31 writes the sound setting and volume selected in the memory 34 (step S404).

[0027] Assume that the user has not selected the sound mode, i.e., has selected the received speech mode also available with the speaker 4 (NO,

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step S402). Then, the CPU 31 controls the speech output switching controller 36 such that a received speech will be output from the speaker 4 with the speech volume selected beforehand (step S405). Specifically, the switching controller 36 turns on the switch 311 and turns off the switch 312 such that the output of the CODEC 35 is fed to the speaker 4 via the received speech amplifier 37. Further, the switching controller 36 controls the resistance of the volume 321 in order to cause the speaker 4 to output a received speech with the speech volume selected beforehand. The CPU 31 writes the sound setting and volume selected in the memory 34 (step S406).

[0028] In the illustrative embodiment, when the cellular phone is in a standby state, the switch 313 is turned on while the resistance of the volume 323 is set at a value corresponding to the preselected call incoming tone volume. In this condition, the output of the CODEC 35 is fed to the speaker 5 via the sound amplifier 39.

[0029] A procedure to follow the termination of a call will be described with reference to FIG. 5. As shown, assume that the radio section 32 detects a call terminated at the cellular phone (YES, step S501). Then, the radio section 32 delivers a call termination signal to the CPU 31. The CPU 31, detected the call termination signal (YES, step S501), reads the sound setting out of the memory 34 and then determines whether or not the sound mode available with the speaker 4 is selected (step S502).

[0030] Assume that the sound mode is selected (YES, step S502). Then, the route from the CODEC 35 to the two sound amplifiers 38 and 39 has already been set up while the call incoming tone volume selected beforehand has already been assigned to the speakers 4 and 5. In this case, the CPU 31 causes the speech output switching controller 36 to vary the resistance of the volume 322 at preselected intervals such that the volume of the speaker 4 approaches the volume (call incoming tone volume) of the speaker 5 stepwise.

[0031] The CPU 31 causes the DSP 33 to read call incoming tone data out of the memory 34 and causes it to generate a call incoming signal in a stereophonic fashion. Alternatively, the CPU 31 may cause the DSP 33 to generate the call incoming signal in a dual monaural fashion such that the same sound is output from both of the speakers 4 and 5. The DSP 33 reads the call incoming tone data out of the memory 34, generates a call incoming signal, and feeds the call incoming signal to the CODEC 35. The CODEC 35 converts the digital call incoming signal to an analog signal and delivers the analog signal to the speech output switching controller 36.

[0032] The speech output switching controller 36 controls the resistance of the volume 322 step wise in order to increase the volume of sound to be output from the speaker 4 stepwise (step S503), as instructed by the CPU 31. While the call incoming tone is output from both of the speakers 4 and 5, the volume of the tone output from the speaker 4 increases stepwise to the call incoming tone volume selected beforehand (step S504). This prevents the call incoming tone from being abruptly output from the speaker or receiver 4 with a great volume and thereby frees the user from unpleasantness.

[0033] Assume that the received speech mode available with the speaker 4

is selected (NO, step S502). Then, the CPU 31 causes the speech output switching controller 36 to turns off the switch 311 so as to prevent the call incoming tone from being output from the speaker 4. More specifically, in this case, the route from the CODEC 35 to the sound amplifier 39 has already been set up. Consequently, the output signal of the DSP 33 is delivered only to the speaker 5, i.e., the call incoming tone is output only from the speaker 5 (step S511). This informs the user of the call terminated at the phone.

[0034] The user, recognized the call, presses a particular key, i.e., a conversation start key positioned on the operation panel 7 for thereby off-hooking the phone (step S505). In response, the CPU 31 causes the DSP 33 to stop outputting the call incoming tone from the speakers 4 and 5. Further, the CPU 31 causes the speech output switching controller 36 to restore the group of switches 310 to the condition matching with the sound setting stored in the memory 34.

[0035] Subsequently, the CPU 31 reads the sound setting out of the memory 34 and then determines whether or not the sound mode available with the speaker 4 is selected (step S506). If the sound mode is selected (YES, step S506), then the speech output switching controller 36 has already set up the route from the CODEC 35 to the two sound amplifiers 38 and 39. The CPU 31 therefore causes the switching controller 36 to output a speech from the speaker 4 with the speech volume selected. More specifically, the switching controller 36 turns on the switch 311 and turns off the switches 312 and 313 in order to deliver the output signal or received speech signal of the CODEC 35 only to the received speech amplifier 37 (step S507).

[0036] Further, the CPU 31 cause the DSP 33 to process a speech signal received via the radio section 32 and feeds the processed speech signal to the CODEC 35. The CODEC 35 converts the speech signal to an analog speech signal and feeds the analog speech signal to the speech output switching controller 36. The analog speech signal is input to the speaker 4 via the switch 311 of the switching controller 36 and received speech amplifier 37 because the switches 312 and 313 have been turned off.

[0037] The microphone 6 transforms the user's voice input thereto to a speech signal. The microphone amplifier 41 amplifies the speech signal and feeds the amplified speech signal to the CODEC 35. The CODEC 35 digitizes the speech signal and delivers a digital speech signal to the DSP 33. The DSP 33 processes the digital speech signal and feeds the processed digital speech signal to the radio section 32. The radio section 32 sends the processed digital speech signal to a base station. In this manner, conversation is held between the user and the other party (step S508).

[0038] Assume that the sound setting is not the sound mode available with the speaker 4 (NO, step S506). Then, the CPU 31 causes the speech output switching controller 36 to turn off the switch 313. In this case, the route from the CODEC 35 to the received speech amplifier 37 has already been set up. Therefore, the output signal or received speech signal of the CODEC 35 is input to the speaker 4 via the received speech amplifier 37. Consequently, the received speech is output only from the speaker 4, setting up conversation (step S508). After the conversation has been set

up, the illustrative embodiment causes the speech output switching controller 36 to turn off the switch 313, as stated above. This control over the switch 313 is not necessary if the CODEC 35 is configured to output the received speech signal on only one output thereof, which is connected to the switches 311 and 312.

[0039] On completing conversation, the user presses a particular key, e.g., a conversation end key for thereby on-hooking the phone (step S509). In response, the CPU 31 causes the speech output switching controller 36 to restore the group of switches 310 to the condition matching with the sound setting stored in the memory 34 (step S510). Thereafter, the procedure returns to the step S501.

[0040] A procedure for reproducing music data stored in the memory 34 will be described hereinafter. When the phone is connectable to, e.g., Internet, it is capable of receiving music data via Internet. Music data received is written to the memory 34 in the same manner as call incoming tone data. Today, even music data distribution services are available at, e.g., convenience stores and record stores for recording various kinds of music data in MDs (Mini Disks) or similar disks. Even the cellular phone is capable of receiving music data at such a spot if provided with a function of downloading the music data, as will be described hereinafter.

[0041] To reproduce the music data stored in the memory 34, the user presses the menu key or similar preselected key on the operation panel 7. In response, the CPU 31 displays various kinds of selection menus on the display 8. When the user selects a music play menu, the CPU 31 designates music data to be reproduced in accordance with the user's selection.

[0042] Further, by varying the sound setting, the user can enjoy music based on the music data and output from both of the speakers 4 and 5. For example, assume that the user switches the sound setting to the sound mode available with the speaker 4. Then, the volume of the speaker 4 is control led to the volume of the speaker 5 while the output signal of the CODEC 35 is delivered to the speakers 4 and 5 via the sound amplifiers 38 and 39, respectively. When the user performs a preselected operation for reproducing the music, the CPU 31 causes the DSP 33 to reproduce the designated music data. More specifically, the DSP 33 reads the music data out of the memory 34 and generates a music signal in the same manner as when generating the call incoming tone signal. The music signal is delivered to the speakers 4 and 5 via the CODEC 35, speech output switching controller 36, and sound amplifiers 38 and 39. As a result, sound is output form both of the speakers 4 and 5 with the call incoming tone volume in accordance with the music signal.

[0043] At the time of music reproduction, the CPU 31 does not cause the switching controller 36 to increase the volume of the speaker 4 stepwise. Therefore, sound is immediately output from the speaker 4 with the call incoming tone volume selected beforehand. The user may select a greater call incoming tone volume to enjoy the music with a greater volume, if desired.

[0044] The music output from both of the speakers 4 and 5 provides the user with a stereophonic effect. It is to be noted that when the user inputs a

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music play command on the operation panel 7, the CPU 31 may automatically switch the sound setting to the sound mode available with the speaker 4.

[0045] While the illustrative embodiment reproduces music data by using the sound setting, it may be modified to reproduce music data without regard to the sound setting, if desired. For example, the phone may be configured to allow the user to select a desired sound volume independently of the call incoming tone volume, in which case the phone will reproduce music data with the desired sound volume in place of the preselected call incoming tone volume. Further, the phone may be configured to allow the user to select the speaker 5, the speakers 4 and 5 or the speaker 5 for the reproduction of music data. In addition, the user, when selected the speaker 4 for the reproduction of music data, may select the speech volume so as to enjoy music in the same manner as usual conversation.

[0046] As stated above, the phone includes the speakers 4 and 5 each of which can output sound with a volume greater than the speech volume. The phone can therefore output a call incoming tone or music with a stereophonic effect. Particularly, the stereophonic effect is enhanced because the speakers 4 and 5 are spaced from each other on the same surface that faces the front case 1.

[0047] Further, the user can enjoy stereophonic music only if a speaker conventionally mounted on a cellular phone is replaced with the speaker 4 shared by speeches and sound. This can be done without resorting to any extra audio part or any extra mounting space. The illustrative embodiment can therefore output stereophonic sound despite its size equivalent to the size of a conventional cellular phone.

[0048] Moreover, the illustrative embodiment causes a call incoming tone to be output from the speaker 4 with a volume increasing stepwise, i.e., prevents a call incoming tone from being abruptly output with a great volume. The user is therefore free from unpleasantness.

[0049] In summary, in accordance with the present invention, a cellular phone includes a first speaker capable selectively outputting a received speech or sound and a second speaker capable of outputting sound. A controller causes sound to be selectively output from one or both of the first and second speakers in accordance with sound setting selected by the user of the phone. The phone of the present invention can therefore output stereophonic sound, e.g., a stereophonic call incoming tone. Further, the first speaker can be implemented by the speaker of a conventional cellular phone only if a function of outputting sound, as distinguished from a speech, is added to the speaker. The phone of the present invention can therefore output stereophonic sound with a size and a weight comparable with those of the conventional cellular phone.

[0050] Various modifications will become possible for those skilled in the art after receiving the teachings of the present disclosure without departing from the scope thereof.

Portable telephone set

Claims of corresponding document: **US2002042287**

What is claimed is:

1. A cellular phone for interchanging information with a base station included in a mobile communication system by radio, said cellular phone comprising:
a first speaker for selectively outputting a received speech or sound;
a second speaker for outputting sound; and
a controller for controlling output of a received speech or sound from said first speaker and said second speaker in accordance with sound setting selected beforehand.

2. The phone as claimed in claim 1, wherein the sound setting is to cause said first speaker to output a received speech or to cause said first speaker to output sound.

3. The phone as claimed in claim 2, further comprising:
a received speech amplifier and a sound amplifier connected to said first speaker for amplifying a received speech and sound, respectively; and
a sound amplifier connected to said second speaker for amplifying sound.

4. The phone as claimed in claim 3, wherein when said first speaker is assigned to a received speech, said controller causes a call incoming signal to be input only to said sound amplifier connected to said second speaker.

5. The phone as claimed in claim 4, wherein during conversation, said controller causes a received speech signal to be input only to said sound amplifier connected to said first speaker.

6. The phone as claimed in claim 5, further comprising a speech memory for storing speech data, wherein said controller causes, when said speech data should be reproduced, a speech signal to be input to said sound amplifier connected to said first speaker and said sound amplifier connected to said second speaker.

7. The phone as claimed in claim 6, further comprising:
a radio section for interchanging information with a remote station via an antenna;
a memory for storing data;
a signal processor for executing preselected processing with a signal received via said radio section or reading data out of said memory to thereby generate a signal corresponding to said data; and
a digital-to-analog converter for digitizing an output signal of said signal processor and delivering a resulting digital signal to said controller.

8. The phone as claimed in claim 3, wherein when said first speaker is assigned to sound, said controller causes a call incoming tone signal to be fed to said sound amplifier connected to said first speaker and said sound amplifier connected to said second speaker.

9. The phone as claimed in claim 8, wherein said controller causes a volume of the call incoming signal output from said first speaker to increase stepwise

10. The phone as claimed in claim 1, further comprising:
a received speech amplifier and a sound amplifier connected to said first speaker for amplifying a received speech and sound, respectively; and
a sound amplifier connected to said second speaker for amplifying sound.

11. The phone as claimed in claim 10, wherein when said first speaker is assigned to a received speech, said controller causes a call incoming signal to be input only to said sound amplifier connected to said second speaker.

12. The phone as claimed in claim 11, wherein during conversation, said controller causes a received speech signal to be input only to said sound amplifier connected to said first speaker.

13. The phone as claimed in claim 12, further comprising a speech memory for storing speech data, wherein said controller causes, when said speech data should be reproduced, a speech signal to be input to said sound amplifier connected to said first speaker and said sound amplifier connected to said second speaker.

14. The phone as claimed in claim 13, further comprising:
a radio section for interchanging information with a remote station via an antenna;
a memory for storing data;
a signal processor for executing preselected processing with a signal received via said radio section or reading data out of said memory to thereby generate a signal corresponding to said data; and
a digital-to-analog converter for digitizing an output signal of said signal processor and delivering a resulting digital signal to said controller.

15. The phone as claimed in claim 10, wherein when said first speaker is assigned to sound, said controller causes a call incoming tone signal to be fed to said sound amplifier connected to said first speaker and said sound amplifier connected to said second speaker.

16. The phone as claimed in claim 15, wherein said controller causes a volume of the call incoming signal output from said first speaker to increase stepwise

17. The phone as claimed in claim 1, further comprising:

a radio section for interchanging information with a remote station via an antenna;
a memory for storing data;
a signal processor for executing preselected processing with a signal received via said radio section or reading data out of said memory to thereby generate a signal corresponding to said data; and
a digital-to-analog converter for digitizing an output signal of said signal processor and delivering a resulting digital signal to said controller.

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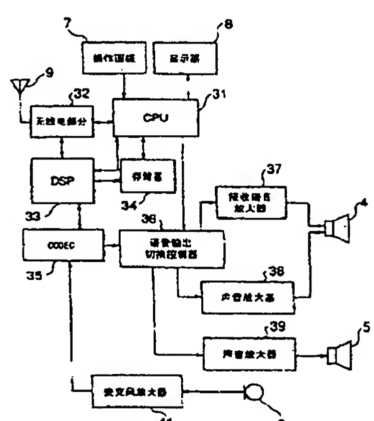
权利要求书3页 说明书9页 附图页数5页

[54]发明名称 便携式电话机

[57]摘要

一种用于通过本发明的无线电与基站交换信息的便携式电话机,包 括用于选择性输出接收的语音或声音的第一扬声器和用于输出声音的第 二扬声器。控制器根据电话机用户预先选择的声音设置控制从第一和第 二扬声器接收的语音或声音的输出。



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权 利 要 求 书

1. 一种用于通过无线电与包括在移动通信系统中的基站交换信息的便携式电话机，所述便携式电话机包括：

第一扬声器，其用于选择性输出接收的语音或声音；

第二扬声器，其用于输出声音；和

控制器，其用于根据预先选择的声音设置控制从所述第一扬声器和所述第二扬声器的接收的语音或声音的输出。

2. 根据权利要求 1 所述的电话机，其特征在于该声音设置将使得所述第一扬声器输出接收的语音或使得所述第一扬声器输出声音。

3. 根据权利要求 2 所述的电话机，其特征在于还包括：

连接于所述第一扬声器的接收语音放大器和声音放大器，用于分别放大接收的语音和声音；和

连接于所述第二扬声器的用于放大声音的声音放大器。

4. 根据权利要求 3 所述的电话机，其特征在于当所述第一扬声器被分配用于接收的语音时，所述控制器使得来话呼叫信号仅被输入到连接于所述第二扬声器的所述声音放大器。

5. 根据权利要求 4 所述的电话机，其特征在于在会话期间，所述控制器使得接收的语音信号仅被输入到连接于所述第一扬声器的所述声音放大器。

6. 根据权利要求 5 所述的电话机，其特征在于还包括用于存储音乐数据的语音存储器，其中当所述音乐数据应该被再现时，所述控制器使得语音信号被输入到连接于所述第一扬声器的所述声音放大器和连接到所述第二扬声器的所述声音放大器。

7. 根据权利要求 6 所述的电话机，其特征在于还包括：

无线电部分，其用于通过天线与远程站交换信息；

用于存储数据的存储器；

信号处理器，其用于执行对通过所述无线电部分接收的信号预选的处理或从所述存储器读出数据以因此产生对应于所述数据的信号；和

数字-模拟转换器，用于数字化所述信号处理器的输出信号并将得出的数字信号传送到所述控制器。

8. 根据权利要求 3 所述的电话机，其特征在于当所述第一扬声器被分配用于声音时，所述控制器使得来话呼叫音调信号被馈送到连接于所述第一扬声器的所述声音放大器和连接到所述第二扬声器的所述声音放大器。

9. 根据权利要求 8 所述的电话机，其特征在于所述控制器使得从所述第一扬声器输出的来话呼叫信号的音量逐渐增加。

10. 根据权利要求 1 所述的电话机，其特征在于还包括：

连接于所述第一扬声器的接收语音放大器和声音放大器，用于分别放大接收的语音和声音；和

连接于所述第二扬声器的用于放大声音的声音放大器。

11. 根据权利要求 10 所述的电话机，其特征在于当所述第一扬声器被分配用于接收语音时，所述控制器使得来话呼叫信号仅被输入到连接于所述第二扬声器的所述声音放大器。

12. 根据权利要求 11 所述的电话机，其特征在于在会话期间，所述控制器使得接收语音信号仅被输入到连接于所述第一扬声器的所述声音放大器。

13. 根据权利要求 12 所述的电话机，其特征在于还包括用于存储音乐数据的语音存储器，其中当所述音乐数据应该被再现时，所述控制器使得语音信号被输入到连接于所述第一扬声器的所述声音放大器和连接到所述第二扬声器的所述声音放大器。

14. 根据权利要求 13 所述的电话机，其特征在于还包括：

无线电部分，其用于通过天线与远程站交换信息；

用于存储数据的存储器；

信号处理器，其用于执行对通过所述无线电部分接收的信号预选的处理或从所述存储器读出数据以因此产生一个对应于所述数据的信号；和

数字-模拟转换器，其用于数字化所述信号处理器的输出信号并将得出的数字信号传送到所述控制器。

15. 根据权利要求 10 所述的电话机，其特征在于当所述第一扬声器被分配用于声音时，所述控制器使得来话呼叫音调信号被馈送到连接于所述第一扬声器的所述声音放大器和连接到所述第二扬声器的所述声音放大器。

16. 根据权利要求 15 所述的电话机，其特征在于所述控制器使得从所述第一扬声器输出的来话呼叫信号的音量逐渐增加。

17. 根据权利要求 1 所述的电话机，其特征在于还包括：

无线电部分，其用于通过天线与远程站交换信息；

用于存储数据的存储器；

信号处理器，其用于执行对通过所述无线电部分接收的信号预选的处理或从所述存储器读出数据以因此产生一个对应于所述数据的信号；和

数字-模拟转换器，其用于数字化所述信号处理器的输出信号并将得出的数字信号传送到所述控制器。

说　　明　　书

便携式电话机

技术领域

本发明涉及一种用于与包括在一个移动通讯系统中的基站交换信息的便携式电话机。

背景技术

现代便携式电话机通常包括用于输出来话呼叫音调和收到的语音的扬声器和用于存储例如来话呼叫音调数据的存储器。更具体地，扬声器中的一个被分配用于来话呼叫音调，同时另一个扬声器被分配用于接收的语音，并且在这个意义上，充当接收器的角色。 对于分配用于声音的扬声器，经常使用能通过一个宽频带输出声音的扬声器用于满足多种来话呼叫音调的需要。

随着便携式电话机可用功能的提高，上述存储器的容量正增加到这样一个程度以至甚至音乐或语音可以被记录在电话机中，这允许电话机用户可以再现通过音乐分配机器或在电话机上经因特网分配的音乐。

为允许便携式电话机的用户在电话机上再现音乐，该电话机被如此配置以至基于音乐数据的音乐从分配用于声音的扬声器输出，并在性能上高于分配用于接收的语音的扬声器。可是，这种设定不提供立体音响效果的音乐。当两个扬声器被安装于该电话机来输出立体音响声音时，它们增加了本应为较小尺寸和重量的电话机的尺寸和重量。

涉及本发明的技术披露于例如日本专利公开号为 1-120159，4-243358，6-37920，10-23115 和 10-233826 中。

发明内容

本发明的一个目的在于提供一种能输出立体音响声音的便携式电话机，同时保持一个小尺寸、轻重量的配置。

本发明的用于通过无线电与基站交换信息的便携式电话机包括：第一扬声器，其用于选择性输出接收的语音或声音；和用于输出声音的第二扬声器。一控制器，其根据电话机用户预先选择的声音设置控制从第一和第二扬声器接收的语音或声音的输出。

附图描述

本发明的上述和其他目的、特点和优点将从下面结合附图的详细描述中变得更加明显：

图 1 是一个说明具体实施本发明的便携式电话机的分解的视图；

图 2 是一个说明包括在该图示的实施例中包括的电路的方框图；

图 3 是说明包括在图 2 的电路中的语音输出切换控制器的特定配置的示意性方框图；

图 4 和 5 是说明该例证性实施例的特定操作的流程图。

优选实施方案

参照附图 1，显示了体现本发明的便携式电话机，其包括一个电路板 3。第一和第二扬声器 4 和 5、一个麦克风 6、一个操作面板 7 和一个显示器 8 分别安装于电路板 3 上。第一扬声器 4 选择性输出接收的语音或包括来话呼叫音调和音乐的声音。第二扬声器 5 仅输出声音。第一和第二扬声器 4 和 5 在电路板 3 的一个主表面上相互分开设置以便实现立体音响效果。扬声器 4 被如此放置以至当该电话机的用户在该电话机上会话时，该扬声器 4 面向用户的耳朵。在会话期间，用户的声音被输入到麦克风 6。显示器 8 用例如一个 LCD（液晶显示器）来实现并显示包括电话号码的文本信息和图形信息。为了输入包括电话号码的各种信息，用户可以操作操作面板 7。用于执行该便携式电话机所要求的各种操作的电子电路也被安排在电路板 3 上。扬声器 4 和 5 在性能上一样。

一个前壳体 1 形成有在位置上分别对应于扬声器 4 和 5 和麦克风 6 的各孔 11、12 和 13。前壳体 1 包括与显示器 8 对应的透明部分。另外，在前壳体 1 上形成一组孔并对应于安排在操作面板 7 上的一组键。一个天线 9 被安装在后壳体 2 上并连接于电路板 3。前壳体 1 和后壳体 2 与

介于其间的电路板 3 合在一起。此刻，带有各种电子器件和电路的电路板 3 的表面面向前壳体 1。

图 2 说明了包括在在例证性实施例中的电子电路。如图所示，该电路包括一个 CPU（中央处理单元）31，用于接收包括来自操作面板 7 的键输入信号的各种信号。CPU 31 分别向显示器 8 和语音输出切换控制器 36 传送显示命令和开关命令。另外，CPU 31 与 DSP（数字信号处理器）或信号处理器 33 和存储器 34 交换信息。起语音存储器作用的存储器 34 可以通过例如一个 EEPROM（电子可擦除可编程存储器）来实现。另外，CPU 31 执行该电话机所要求的各种控制。在该例证性实施例中，CPU 31 和语音输出切换控制器 36 一起构成了一个控制器。

DSP 33 与 CPU 31 和存储器 34 以及与无线电部分 32 和一个 CODEC（编码器/解码器）35 交换信息。CODEC 35 包括一个 AD（模拟到数字）转换器和一个 DA（数字到模拟）转换器。存储器 34 存储包括来话呼叫音调数据的各种数据并能够存储包括通过例如因特网接收的音乐数据的其它数据。无线电部分 32 放大功率并转换频率，以便通过天线 9 与包括在一个移动通信系统中的一个基站通信。DSP 33 更正接收的信号的特性，设置便携式电话机和基站之间的无线电信道，切换会话信道，并处理语音信号。

语音输出切换控制器 36 连接于一个接收语音放大器 37 和声音放大器 38 和 39 以及连接到 CPU 31 和 CODEC 35。切换控制器 36 根据从 CPU 31 输出的命令执行音量控制和放大器切换控制。接收语音放大器 37 和声音放大器 38 连接于第一个扬声器 4，而声音放大器 39 连接于第二扬声器 5。声音放大器 38 和 39 每个都具有比接收语音放大器 37 更大的放大比率。更具体地，通过声音放大器 38 从扬声器 4 输出的声音和通过声音放大器 39 从扬声器 5 输出的声音每个都具有比通过接收语音放大器 37 从扬声器 4 输出的语音更大的音量。注意，接收的语音是指与该例证性实施例的便携式电话机通信的另一个便携式电话机的用户的话音。

麦克风 6 将输入的语音转换为一个语音信号。麦克风放大器 41 放大从麦克风 6 输出的语音信号并将放大的语音信号馈送到 CODEC 35。

图 3 说明了语音输出切换控制器 36 的一个特定配置。如图所示，切

换控制器 36 包括一组开关 311 到 313 和一组音量 321 到 323。开关 311 到 313 和音量 321 到 323 总地分别由标号 310 和 320 表示。音量 321 到 323 是其阻抗在电子控制下可变地电子音量。

开关 311 和 312 在一端被连接于 CODEC 35 的一个输出。音量 321 具有一个连接到开关 311 的另一端的输入并具有一个连接到接收语音放大器 37 的输入端的输出。音量 322 具有一个连接到开关 312 的另一端的输入并具有一个连接到声音放大器 38 的输入端的输出。开关 313 的一端被连接于 CODEC 35 的另一个输出。音量 323 具有一个连接到开关 313 的另一端的输入并具有一个连接到声音放大器 39 的输入端的输出。

切换控制器 36 根据从 CPU 31 输出的命令选择性开或关开关 311 到 313 中的每一个并变化音量 321 到 323 中的每一个的阻抗。当期望立体音响效果时，CODEC 35 从它的两个输出中的每个产生一个特别的信号，当不期望这样一个效果时，它从两个输出端产生相同的信号。

在该例证性实施例中，扬声器 4 以两种不同的模式即接收语音模式和声音模式进行操作。这两个模式之一的选择被称为声音设置。当选择接收语音模式时，使得扬声器用之前设置的语音音量输出接收语音。当选择声音模式时，使得扬声器 4 用预选的来话呼叫音调音量输出声音。对于声音设置，在例如装运（shipment）的时刻设置例如该接收语音模式的特别模式。该电话机的用户可以在该电话机上选择一个期望的语音音量和一个期望的来话呼叫音调音量。另外，用户可以选择一个来话呼叫音调的期望的式样。该语音音量指通过扬声器 4 输出的接收语音的音量，同时来话呼叫音调音量指通过扬声器 5 输出的声音的音量。当用户选择扬声器 4 可用的声音模式时，扬声器 4 被控制为与扬声器 5 同样的音量。在该例证性实施例中，扬声器 4 的音量如后所述逐渐升高到扬声器 5 的音量。来话呼叫音调或声音的最大音量大于接收语音的最大音量。

当该电话机处于待机状态或当进行会话或音乐再现时，用户可以执行声音设置并选择一个期望的语音音量和一个期望的来话呼叫音调音量。下面将参照图 4 及图 3 说明该例证性实施例的特定的操作。当用户按压例如位于操作面板 7 上的菜单键的特别键时，CPU 31 在显示器 8 上显示一个菜单。如图 4 中所示，观看菜单画面的用户通过例如移动光标

选择“声音设置”（步骤 S401）。该用户可以额外地选择一个期望的语音音量和一个期望的来话呼叫音调音量。操作面板 7 将用户的选择结果传送到 CPU 31。作为响应，CPU 31 确定用户是否已经选择了扬声器 4 可用的声音模式（步骤 S402）。

假设用户已经选择了声音模式（是，步骤 S402），然后，CPU 31 通过馈送一个命令给切换控制器 36 使得语音输出切换控制器 36 选择一个来话呼叫音调（步骤 S403）。作为响应，该切换控制器 36 关闭开关 311 并打开开关 312 以因此将 CODEC 35 的输出通过声音放大器 38 传送到扬声器 4。另外，为了实现预先选择的来话呼叫音调音量，该切换控制器 36 控制音量 322 的阻抗。CPU 31 将声音设置和选择的音量写入存储器 34 中（步骤 S404）。

假设用户未选择声音模式，即已经选择了扬声器 4 也可用的接收语音模式（否，步骤 S402）。然后，CPU 31 控制语音输出切换控制器 36 以便以预先选择的语音音量从扬声器 4 输出接收语音（步骤 S405）。具体地，该切换控制器 36 打开开关 311 并关闭开关 312 以便 CODEC 35 的输出通过接收语音放大器 37 被馈送到扬声器 4。另外，为了使得扬声器 4 用预先选择的语音音量输出接收的语音，切换控制器 36 控制音量 321 的阻抗。CPU 31 将声音设置和选择的音量写入存储器 34 中（步骤 S406）。

在该例证性实施例中，当便携式电话机处于待机状态时，在对应于预选来话呼叫音调音量的值处打开开关 313，同时设置音量 323 的阻抗。在这种情况下，CODEC 35 的输出通过声音放大器 39 馈送到扬声器 5。

下面参照图 5 描述接着呼叫终止的过程。如所示，假设无线电部分 32 检测到在便携式电话机终止的呼叫（是，步骤 S501）。然后，无线电部分 32 将呼叫终止信号传送到 CPU 31。检测到该呼叫终止信号（是，步骤 S501）的 CPU 31 将声音设置读出存储器 34，然后确定是否选择了扬声器 4 可用的声音模式（步骤 S502）。

假设选择了声音模式（是，步骤 S502）。然后，从 CODEC 35 到两个声音放大器 38 和 39 和路径已经被设置，同时预先选择的来话呼叫音调音量已经被分配给扬声器 4 和 5。在这种情况下，CPU 31 使得语音输出切换控制器 36 以预选的间隔变化音量 322 的阻抗以便扬声器 4 的音量逐

渐接近扬声器 5 的音量（来话呼叫音调音量）。

CPU 31 使得 DSP 33 从存储器 34 读出来话呼叫音调数据并使得它以立体音响的方式产生一个来话呼叫信号。另外，CPU 31 可以使得 DSP 33 以双单音响方式产生来话呼叫信号以便从扬声器 4 和 5 输出相同的声音。DSP 33 从存储器 34 读出该来话呼叫音调数据，产生一个来话呼叫信号，并将该来话呼叫信号馈送到 CODEC 35。CODEC 35 将数字来话呼叫信号转变为模拟信号并将该模拟信号传送给语音输出切换控制器 36。

为了逐渐增加从扬声器 4 输出的声音的音量，在 CPU 31 的指令下，语音输出切换控制器 36 逐渐控制音量 322 的阻抗（步骤 S503）。当来话呼叫音调从扬声器 4 和 5 两者输出时，从扬声器 4 输出的音调的音量逐渐增加到预先选择的来话呼叫音调音量（步骤 S504）。这防止了来话呼叫音调从具有大音量的该扬声器或扬声器 4 被断续地输出并因此免除了用户的不愉快。

假设选择了扬声器 4 可用的接收语音模式（否，步骤 S502）。然后 CPU 31 使得语音输出切换控制器 36 关闭开关 31 以便防止来话呼叫音调从扬声器 4 输出。更具体地，在这种情况下，从 CODEC 35 到声音放大器 39 的路径已经被设置。因此，DSP 33 的输出信号仅被传送到扬声器 5，即来话呼叫音调仅从扬声器 5 输出（步骤 S511）。这通知了在该电话机端头呼叫的用户。

识别该呼叫的用户按压一个特别的键，即位于操作面板 7 上的会话开始键，用于接听（off-hooking）该电话（步骤 S505）。作为响应，CPU 31 停止从扬声器 4 和 5 输出来话呼叫音调。进一步，CPU 31 使得语音输出切换控制器 36 将该组开关 310 恢复为与存储在存储器 34 中的声音设置相匹配的状况。

接下来，CPU 31 从存储器 34 读出该声音设置并然后是否选择了扬声器 4 可用的声音模式（步骤 S506）。如果选择了声音模式（是，步骤 S506），然后，语音输出切换控制器 36 已经设置了从 CODEC 35 到两个声音放大器 38 和 39 的路径。CPU 31 因此使得切换控制器 36 用选择的语音音量从扬声器 4 输出语音。更具体地，切换控制器 36 打开开关 311 并关闭开关 312 和 313 以便将 CODEC 35 的输出信号或接收语音信号仅传送到接收

语音放大器 37（步骤 S507）。

更进一步，CPU 31 使得 DSP 33 处理通过无线电部分 32 接收的语音信号并将处理的语音信号馈送给 CODEC 35。CODEC 35 将该语音信号转变为模拟语音信号并将模拟语音信号馈送给语音输出切换控制器 36。因为已经关闭了开关 312 和 313，模拟语音信号通过切换控制器 36 的开关和接收语音放大器 37 被输入到扬声器 4。

麦克风 6 将输入的用户的话音转变为语音信号。麦克风放大器 41 放大该语音信号并将放大的语音信号馈送给 CODEC 35。CODEC 35 数字化该语音信号并将数字语音信号传送给 DSP 33。DSP 33 处理该数字语音信号并将该处理的数字语音信号馈送给无线电部分 32。无线电部分 32 将该经过处理的数字语音信号发送给一个基站。用这种方式，在用户和另一方之间保持会话（步骤 S508）。

假设该声音设置不是扬声器 4 可用的声音模式（否，步骤 S506）。然后，CPU 31 使得语音输出切换控制器 36 关闭开关 313。在这种情况下，从 CODEC 35 到接收语音放大器 37 的路径已经被设置。因此，CODEC 35 的输出信号或接收语音信号通过接收语音放大器 37 被输入到扬声器 4。因此，接收语音仅从设置会话的扬声器 4 输出（步骤 S508）。在已经设置了会话之后，该例证性实施例如上所述，使得语音输出切换控制器 36 关闭开关 313。如果将 CODEC 35 配置为仅输出接收语音信号到被连接到开关 311 和 312 的一个输出端上，则不需要这个对开关 313 的控制。

当完成会话时，用户按压一个特别键，例如会话结束键用于挂断（on-hooking）该电话（步骤 S509）。作为响应，CPU 31 使得语音输出切换控制器 36 将该组开关 310 恢复为与存储在存储器 34 中的声音设置相匹配的状况(步骤 S510)。因此，该过程返回到步骤 S501。

下面将描述用于再现存储在存储器 34 中的音乐数据的过程。该电话机可以连接到例如因特网时，它能够通过因特网接收音乐数据。接收的音乐数据以与来话呼叫音调数据相同的方式被写入存储器 34。今天，甚至音乐数据分配服务可以在例如用于将各种音乐数据记录在 MD（迷你盘）或其它类似盘上的用具商店和唱片商店处获得。如后面将描述的，如果提供有下载音乐数据的功能，甚至便携式电话机能够在这样的地点接收

音乐数据。

为了再现存储在存储器 34 中的音乐数据，用户按压操作面板 7 上的菜单键或类似的预选键。作为响应，CPU 31 在显示器 8 上显示各种选择菜单。当用户选择了音乐播放菜单时，CPU 31 根据用户的选择指定将被再现的音乐数据。

进一步，通过改变声音设置，用户可以根据音乐数据欣赏音乐并从扬声器 4 和 5 两者输出。例如，假设用户将声音设置切换为扬声器 4 可用的声音模式。然后控制扬声器 4 的音量到扬声器 5 的音量，同时 CODEC 35 的输出信号通过声音放大器 38 和 39 分别被传送到扬声器 4 和 5。当用户执行用于再现音乐的预选操作时，CPU 31 使得 DSP 33 再现该指定的音乐数据。更具体地，DSP 33 从存储器 34 读出音乐数据并以与产生来话呼叫音调信号相同的方式产生音乐信号。该音乐信号通过 CODEC 35、语音输出切换控制器 36 和声音放大器 38 和 39 传送给扬声器 4 和 5。结果，声音按照该音乐信号用来话呼叫音调音量从扬声器 4 和 5 两者输出。

在音乐再现的时刻，CPU 31 不使切换控制器 36 逐渐增加扬声器 4 的音量。因此，声音立即用预先选择的来话呼叫音调音量从扬声器 4 输出。如果期望，用户可以选择更大的来话呼叫音调音量以用更大的音量欣赏音乐。

从扬声器 4 和 5 两者输出的音乐提供给用户立体音响的效果。注意，当用户在操作面板 7 上输入音乐播放命令时，CPU 31 可以自动将声音设置切换为扬声器 4 可用的声音模式。

当该例证性实施例通过使用声音设置再现音乐数据时，如果期望的话，它也可以被修改为与声音设置无关地再现音乐数据。例如，该电话机可以被配置为允许用户选择独立于来话呼叫音调音量的所期望的声音音量，在这种情况下，该电话机将用期望的声音音量再现音乐数据而不用预选的来话呼叫音调音量。进一步，该电话机可以被配置为允许用户选择扬声器 5、扬声器 4 和 5 或扬声器 5 用于音乐数据的再现。另外，当选则扬声器 4 用于音乐数据的再现时，用户可以选择语音音量以便以与通常会话相同的方式欣赏音乐。

如上所述，该电话机包括扬声器 4 和 5，其中每一个可以用大于语音

音量的音量输出声音。该电话机因此可以用立体音响的效果输出一个来话呼叫音调。具体地，因为扬声器 4 和 5 在面向前壳体 1 的相同表面上相互分开放置，该立体音响的效果得到强化。

进一步，如果常规安装于一个便携式电话机上的扬声器用由语音和声音共享的扬声器 4 替换掉时，用户可以欣赏立体音响的音乐。这可以不用借助于任何特殊的音频部件或任何特殊的安装空间来实现。因此，不管它的尺寸是否相当于常规便携式电话机的尺寸，该例证性实施例可以输出立体音响的声音。

再有，该例证性实施例使得来话呼叫音调用逐渐增加的音量从扬声器 4 输出，即防止来话呼叫音调用大音量断续地输出。用户因此可以免除不愉快。

总地来说，根据本发明，便携式电话机包括能够选择性输出接收语音或声音的第一扬声器和能够输出声音的第二扬声器。一个控制器使得声音根据由电话机的用户选择的声音设置，从第一或第二扬声器的一个或两者选择性地输出。本发明的电话机因此可以输出例如立体音响来话呼叫音调的立体音响声音。进一步，仅仅如果作为与语音的区别，输出声音的功能被添加到该扬声器，第一扬声器才可以通过常规便携式电话机的扬声器来实现。本发明的电话机因此可以用可与常规便携式电话机的尺寸和重量相比较的尺寸和重量输出立体音响的声音。

在接受本发明公开的授意后，在不脱离本发明的范围的情况下，本领域的技术人员可能做出各种修改。

说 明 书 附 图

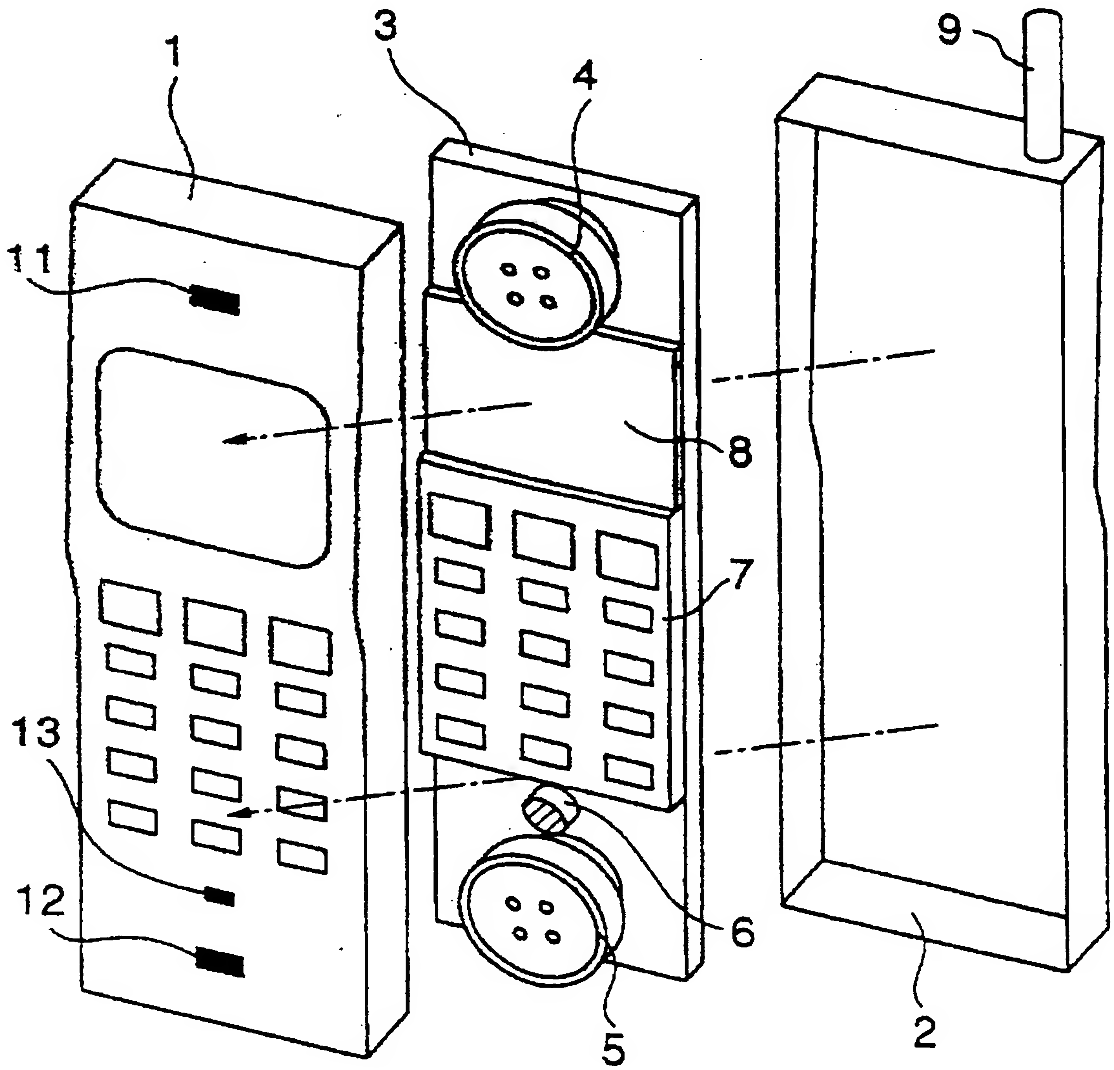
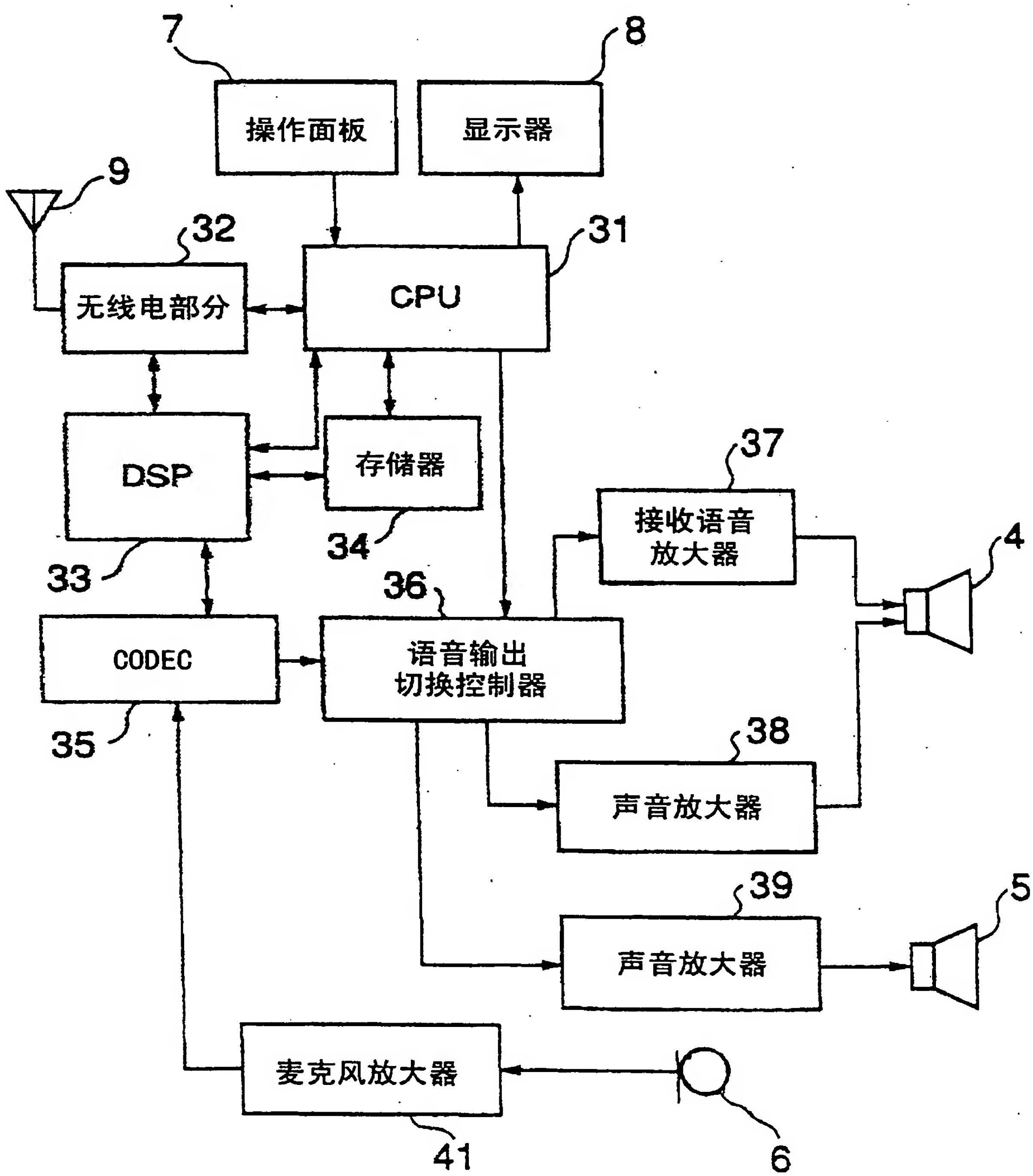


图 1

01.10.11



7
8
操作面板
显示器
9
32
31
无线电部分
CPU
DSP
存储器
37
34
接收语音
放大器
4
33
36
CODEC
语音输出
切换控制器
35
38
声音放大器
39
5
声音放大器
麦克风放大器
41
6

图 2

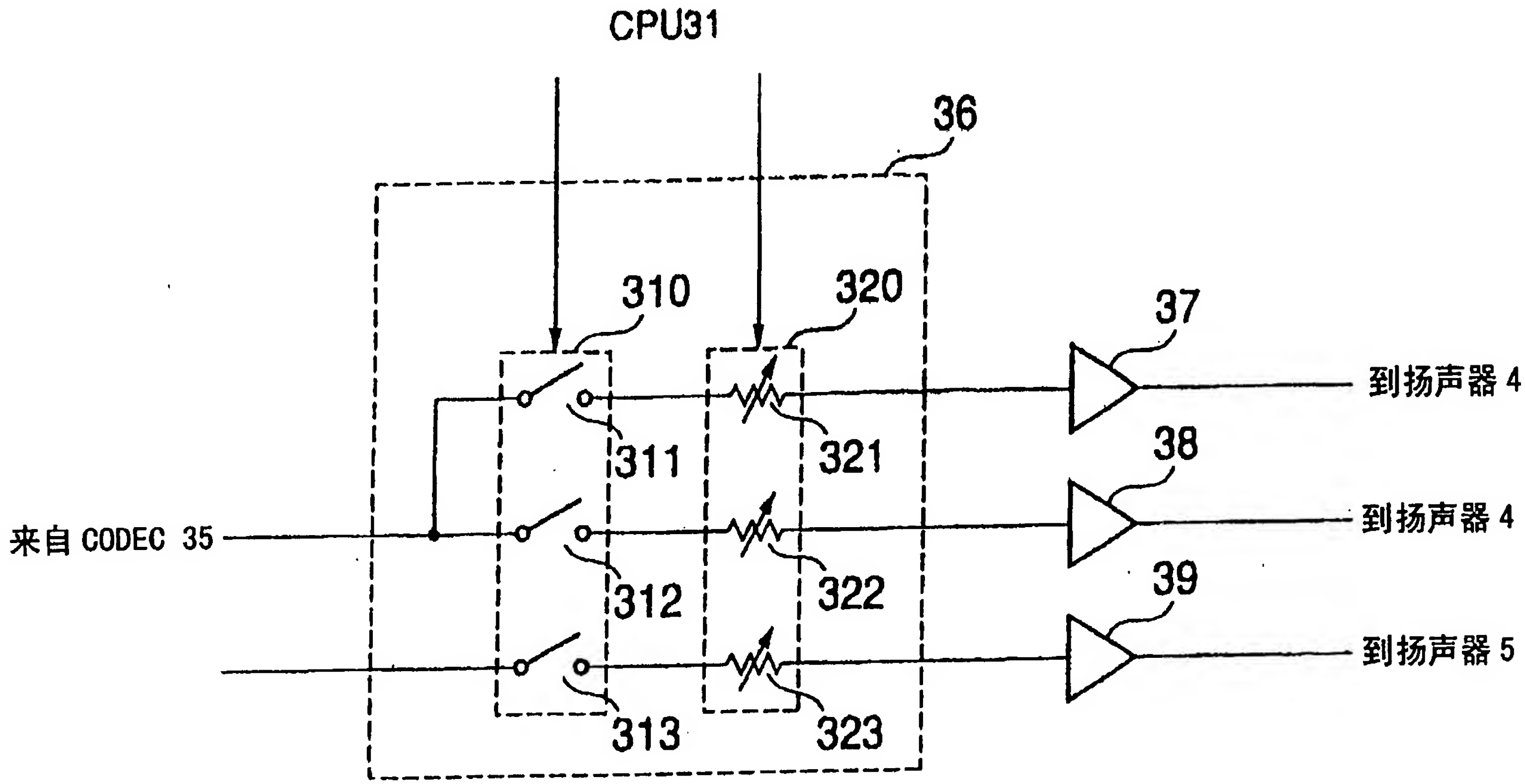

CPU31
36
310
320
37
到扬声器 4
311
321
38
来自 CODEC 35
到扬声器 4
312
322
39
到扬声器 5
313
323

图 3

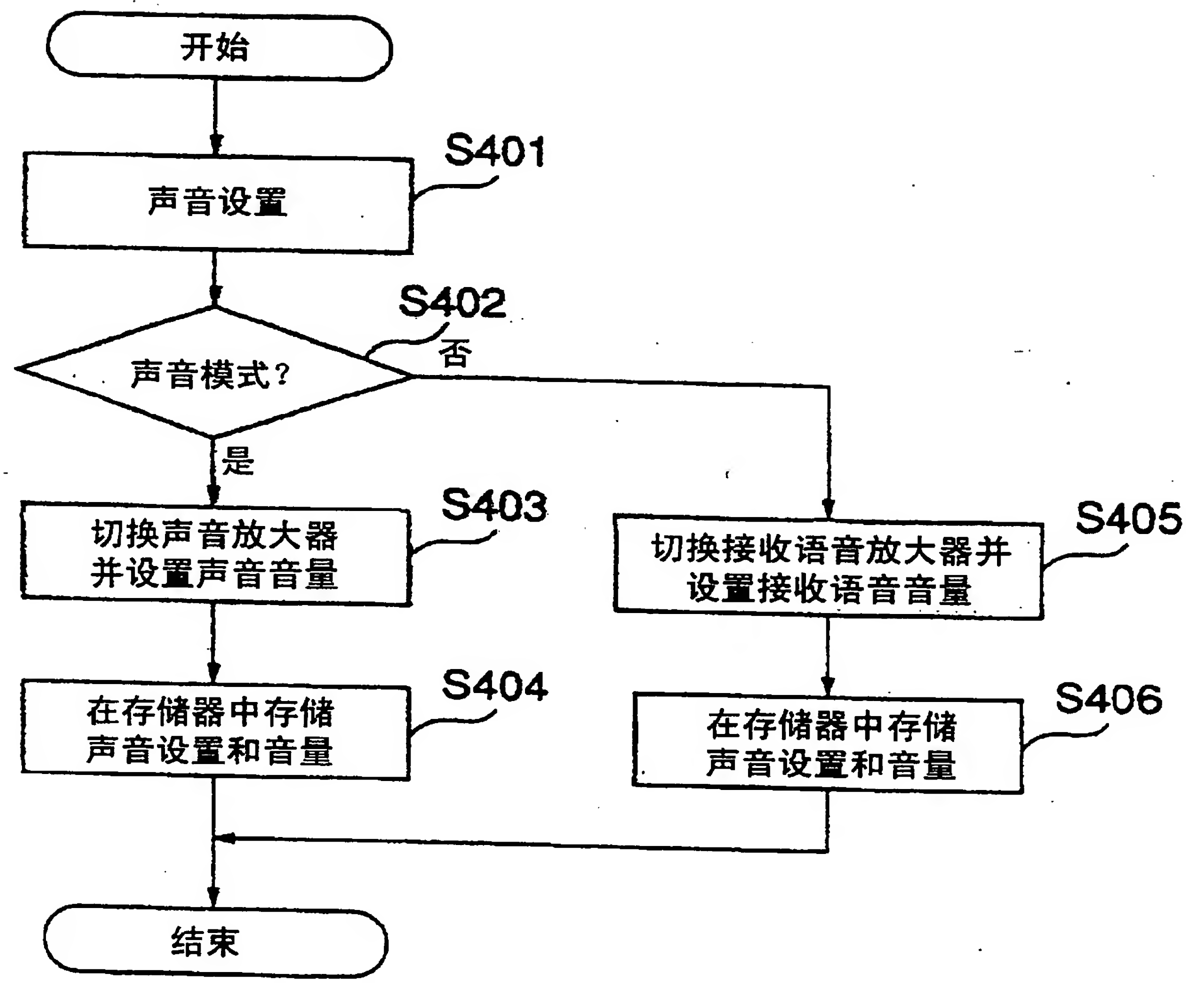

开始
S401
声音设置
S402
声音模式？
否
是
S403
切换声音放大器
并设置声音音量
S405
切换接收语音放大器并
设置接收语音音量
S404
在存储器中存储
声音设置和音量
S406
在存储器中存储
声音设置和音量
结束

图 4

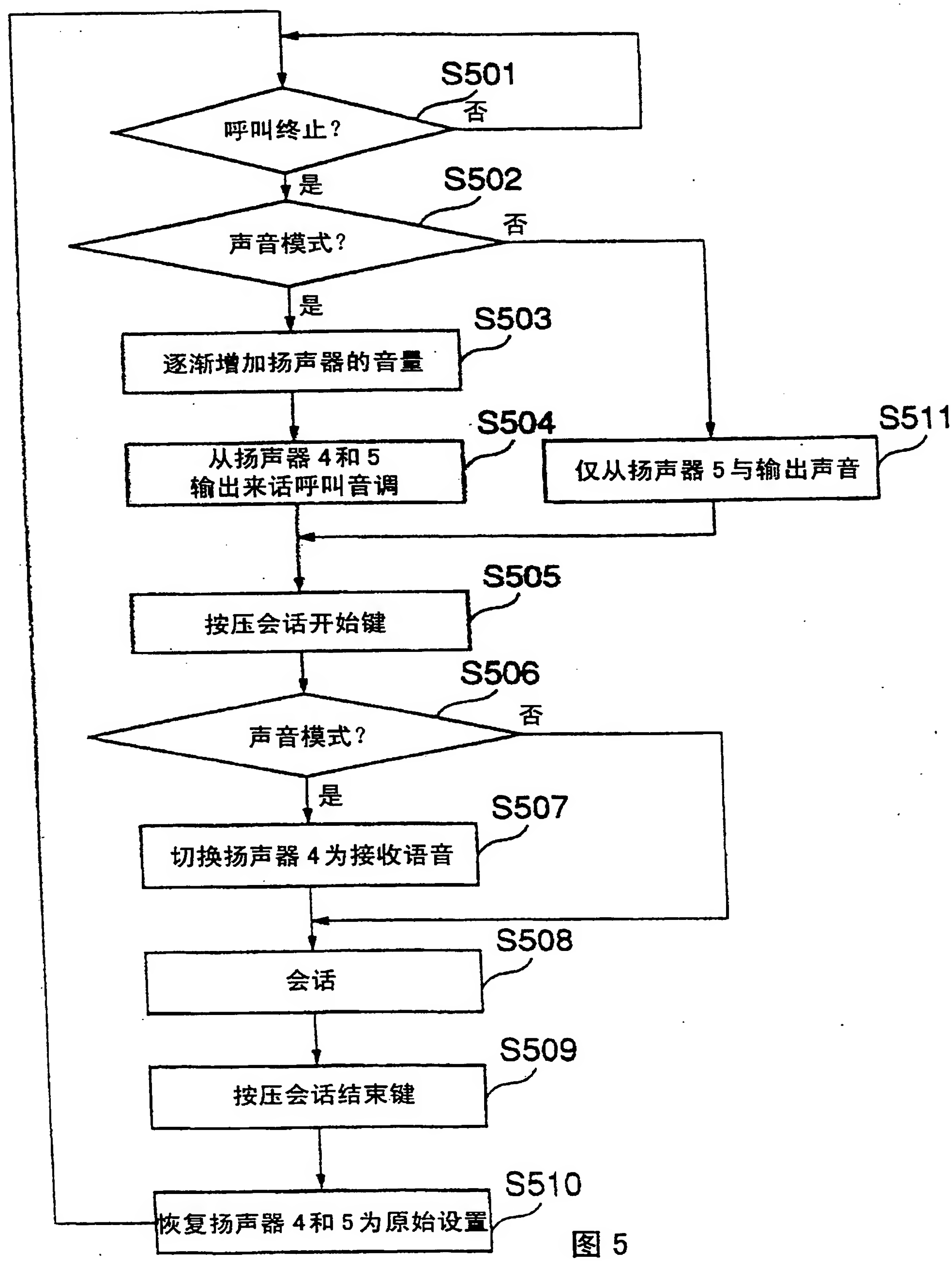

S501
呼叫终止？
否
是
S502
声音模式？
否
是
S503
逐渐增加扬声器的音量
S504
从扬声器 4 和 5
输出来话呼叫音调
S511
仅从扬声器 5 与输出声音
S505
按压会话开始键
S506
声音模式？
否
是
S507
切换扬声器 4 为接收语音
S508
会话
S509
按压会话结束键
S510
恢复扬声器 4 和 5 为原始设置

图 5